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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/388,010	09/01/1999	RUSSELL H. LAMBERT	15-0195	3282

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EXAMINER	
ARMSTRONG, ANGELA A	
ART UNIT	**PAPER NUMBER**
2654	18

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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No. 09/388,010	Applicant(s) LAMBERT ET AL.	
	Examiner Angela A. Armstrong	Art Unit 2654	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *17 October 2003*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *11-33* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *11-33* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTION

Continued Examination Under 37 CFR 1.114

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on October 17, 2003 has been entered.

Claim Rejections - 35 USC § 103

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims 11-13 and 16-18 are rejected under 35 U.S.C. 103(a) as being unpatentable over Marash (US Patent No. 5,825,898) in view of Coker (US Patent No. 4,581,758).

4. Marash teaches a system and method for adaptive interference canceling for reducing interference in a signal received from an array of sensors.

5. Regarding claims 11 and 16, at col. 4, lines 50-67 Marash teaches a sensor array having individual sensors which receives signals from a signal source and from interference sources

(which reads on a plurality of microphones positioned to detect speech from a single speech source and noise from multiple sources), producing a main channel representing signals received in the direction of the source, such that the main channel contains both a source signal component and interference signal component (which reads on generating microphone output signals, such that the reference microphone receives acoustic signals both from the speech source and from the multiple noise sources) producing a reference channel representing signals from directions other than the that of the signal source.

Marash does not specifically teach a plurality of bandpass filters for eliminating spectral bands containing noise from the microphone output signals. Coker et al teach a system of acoustic direction identification of a desired sound source in a noisy reverberant environment. Specifically Coker et al teach implementation of bandpass filters for removing low-frequency components of speech and for eliminating high frequency noise, which can produce unwanted spurious events (col. 3, lines 34-41).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the adaptive interference canceling system of Marash to implement bandpass filtering of the input signal as suggested by Coker et al, for the purpose of eliminating high frequency noise which can produce unwanted spurious events, as taught by Coker et al.

At col. 8, lines 23-67, Marash teaches a plurality of adaptive filters to process the output signals from the reference channels with the output signals from the main channel, which reads on the plurality of adaptive filters, for aligning each data microphone output signal with the output signal from the reference microphone.

At col. 6, lines 1-27, Marash teaches the processing of the main channel matrix to produce a main channel as a weighted sum of outputs which filters a signal coming in all directions to produce a signal coming in a specific direction, which reads on the signal summation circuit for combining filtered output signals from the microphones whereby signal components resulting from speech are combined coherently and signal components from noise combine incoherently, to produce an increased signal-to-noise ratio.

At col. 6, lines 28-48, Marash teaches the processing of the reference channel matrix to produce a reference channel as a weighted sum of interference signals, which reads on a signal summation circuit for combining filtered output signals from the microphones whereby signal components resulting from noise are combined.

Regarding claims 12 and 17, neither Marash nor Coker teaches speech detection and enabling the adaptation process only when speech is detected. However, it would have been obvious to one of ordinary skill at the time of the invention to modify the adaptive interference reduction system of Marash and Coker to implement enabling the adaptation process only when speech is detected, because such a modification would eliminate unnecessary filter weighting adaptation processing, making the system more efficient and economical.

Regarding claims 13 and 18, at col. 8, lines 46-48, Marash discloses a difference unit for subtracting the reference channel matrix (interference channel) from the main channel matrix (the source and interference channel) to effectively reduce the interference in the signal of interest, which reads on the speech conditioning circuitry coupled to the signal summation circuit to reduce reverberation effects.

6. Claims 14-15 and 19-20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Marash in view of Coker as applied to claims 13 and 18 above, and further in view of the Digital Signal Processing Handbook (1998).

7. Regarding claims 14-15 and 19-20, Marash and Coker do not specifically teach filtering microphone output signals by convolution with a vector of weight values in the frequency domain; comparing filtered data signals from data microphones with reference microphones and deriving an error signal; adjusting or updating weight values convolved with data microphone output signals in the frequency domain to minimize the error signal; converting the weights to the time domain using an inverse fast Fourier transform; zeroing out portions of the filter weight values that give rise to unwanted circular convolution; converting filter values to the frequency domain; and fast Fourier transforms (FFT) to transform blocks of microphone output signals to a frequency domain representation.

The DSP Handbook teaches a block based adaptive filter algorithm and provides a summarization of the frequency domain block LMS, which performs an FFT on input signals, updates weight vector, forces some values to zero, performs an inverse fast Fourier transform and determines an error signal (page 22-15 continuing to page 22-17; page 8-1 continuing to page 8-5). The DSP Handbook teaches the block based LMS algorithm is an efficient adaptive filtering algorithm aimed at increasing convergence speed and reducing computational complexity.

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the system of Marash to implement the frequency domain block LMS, as taught in the DSP Handbook, for the purpose of achieving an efficient adaptive filtering algorithm which

increases convergence speed and reduces computational complexity, to thereby improve system efficiency and performance.

8. Claims 21-33 are rejected under 35 U.S.C. 103(a) as being unpatentable over Marash in view of the Digital Signal Processing Handbook (1998).

9. Regarding claims 21-23, 25, 28-30, and 32, Marash teaches a method for improving detection of speech signals comprising reference microphone data and data microphone data at col. 4, lines 50-67. Marash does not specifically teach filtering microphone output signals by convolution with a vector of weight values in the frequency domain; adjusting or updating weight values; converting the weights to the time domain; zeroing out portions of the filter weight values; and converting values to a frequency domain representation.

The DSP Handbook teaches a block based adaptive filter algorithm and provides a summarization of the frequency domain block LMS, which performs an FFT on input signals, updates weight vector, forces some values to zero, performs an inverse fast Fourier transform and determines an error signal (page 22-15 continuing to page 22-17; page 8-1 continuing to page 8-5). The DSP Handbook teaches the block based LMS algorithm is an efficient adaptive filtering algorithm aimed at increasing convergence speed and reducing computational complexity.

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the system of Marash to implement the frequency domain block LMS, as taught in the DSP Handbook, for the purpose of achieving an efficient adaptive filtering algorithm which

increases convergence speed and reduces computational complexity, to thereby improve system efficiency and performance.

Regarding claims 24 and 31, Marash does not teach updating the filter weight value comprising W(k+1)=W(k)+μ[Ref(k)-X(k)*conj(Y)]. The DSP Handbook teaches a block based adaptive filter algorithm and provides a summarization of the frequency domain block LMS, which performs an FFT on input signals, updates weight vector via equation 22.33 on page 22-17, forces some values to zero, performs an inverse fast Fourier transform and determines an error signal (page 22-15 continuing to page 22-17; page 8-1 continuing to page 8-5). The DSP Handbook teaches the block based LMS algorithm is an efficient adaptive filtering algorithm aimed at increasing convergence speed and reducing computational complexity.

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the system of Marash to implement the frequency domain block LMS with the vector weight updating, as taught in the DSP Handbook, for the purpose of achieving an efficient adaptive filtering algorithm which increases convergence speed and reduces computational complexity, to thereby improve system efficiency and performance.

Regarding claim 26, Marash does not teach zeroing out portions of the weight value zeroes out portions that give rise to circular convolutions. The DSP Handbook teaches zeroing our portions of a signal to cyclic convolution so as to remove aliasing effects (section 8.2, page 8-2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the system of Marash to implement zeroing portions of the signal that are from circular

or cyclic convolutions, as suggested in the DSP Handbook, to remove aliasing effects to thereby improve the signal processing performance.

Regarding claims 27 and 33, Marash teaches a method for improving detection of speech signals comprising reference microphone data and data microphone data at col. 4, lines 50-67. Marash does not specifically teach comparing filtered data signals from data microphones with reference microphones and deriving an error signal; adjusting or updating weight values convolved with data microphone output signals in the frequency domain to minimize the error signal.

The DSP Handbook teaches a block based adaptive filter algorithm and provides a summarization of the frequency domain block LMS, which performs an FFT on input signals, updates weight vector, forces some values to zero, performs an inverse fast Fourier transform and determines an error signal (page 22-15 continuing to page 22-17; page 8-1 continuing to page 8-5). The DSP Handbook teaches the block based LMS algorithm is an efficient adaptive filtering algorithm aimed at increasing convergence speed and reducing computational complexity.

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the system of Marash to implement the frequency domain block LMS, as taught in the DSP Handbook, for the purpose of achieving an efficient adaptive filtering algorithm which increases convergence speed and reduces computational complexity, to thereby improve system efficiency and performance.

Response to Arguments

10. Applicant's arguments filed October 17, 2003 have been fully considered but they are not persuasive. Applicant argues the output after summation in Marash does not teach the same output after summation of applicant's invention. The Examiner disagrees and argues Marash teaches a summation implementation wherein the data output from the main channel and reference channels are combined (Figure 11C, element 260), which reads on "signal summation circuit for combining filtered output signals from the microphones whereby signal components resulting from speech are combined coherently and signal components from noise combine incoherently, to produce an increased signal-to-noise ratio", since the system specifically distinguishes the desired speech data from the undesired noise data to reduce the effects of noise, and thereby producing an increased signal-to-noise ratio.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Angela A. Armstrong whose telephone number is 703-308-6258. The examiner can normally be reached on Monday-Thursday 7:30-5:00 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (703) 305-9645. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

Angela A. Armstrong
Examiner
Art Unit 2654

AAA
February 22, 2004

Angela Armstrong